WHAT IS CLAIMED IS:

1. A speech coding apparatus including at least

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode, and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount

which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

2. A speech coding apparatus including at least

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section

indicates a predetermined mode, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

3. A speech coding apparatus including at least

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a

combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

4. A speech coding apparatus including at least

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the

spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a mode on the basis of a past quantized gain of a adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode, and a gain codebook for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

5. A speech decoding apparatus comprising:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information;

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive

codebook; and

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information when an output from said discrimination section indicates a predetermined mode,

wherein a speech signal is reproduced by passing the sound source signal through a synthesis filter section constituted by spectrum parameters.

6. A speech coding/decoding apparatus comprising:

a speech coding apparatus including

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by

a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source

information when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal.

7. A speech coding/decoding apparatus comprising:

a speech coding apparatus including

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

said sound source quantization section for outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating positions of pulses according to a predetermined rule and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters and reproduces a speech signal by

filtering the sound source signal.

8. A speech coding apparatus comprising:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter;

means for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal; and

mode discrimination means for receiving a past quantized adaptive codebook gain and performs mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a predetermined threshold, and

further comprising:

sound source quantization means for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shifting a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech;

Sub B3 gain quantization means for quantizing a gain by using a gain codebook; and

multiplex means for outputting a combination of outputs from said spectrum parameter calculation means, said adaptive codebook means, said sound source quantization means, and said gain quantization means.

9. An apparatus according to claim 8, wherein said sound source quantization means uses a position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode.

10. An apparatus according to claim 9, wherein when mode discrimination indicates a predetermined mode, a predetermined number of pulse positions are generated by random number generating means and output to said sound source quantization means.

11. An apparatus according to claim 8, wherein when mode discrimination indicates a predetermined mode, said sound source quantization means selects a plurality of combinations from combinations of all code vectors in said codebook and shift amounts for pulse positions in an order in which a predetermined distortion amount is minimized, and outputs the combinations to said gain quantization means, and

said gain quantization means quantizes a plurality of sets of outputs from said sound source quantization means

by using said gain codebook, and selects a combination of a shift amount, sound source code vector, and gain code vector which minimizes the predetermined distortion amount.